US006912178B2

(12) **United States Patent**
Chu et al.

(10) **Patent No.: US 6,912,178 B2**
(45) **Date of Patent: Jun. 28, 2005**

(54) **SYSTEM AND METHOD FOR COMPUTING A LOCATION OF AN ACOUSTIC SOURCE**

(75) Inventors: **Peter L. Chu**, Lexington, MA (US); **Michael Kenoyer**, Austin, TX (US); **Richard Washington**, Marble Falls, TX (US)

(73) Assignee: **Polycom, Inc.**, Milpitas, CA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **10/414,421**

(22) Filed: **Apr. 15, 2003**

(65) **Prior Publication Data**

US 2004/0032796 A1 Feb. 19, 2004

Related U.S. Application Data

(60) Provisional application No. 60/372,888, filed on Apr. 15, 2002.

(51) **Int. Cl.**[7] **G01S 3/80**

(52) **U.S. Cl.** **367/123**

(58) **Field of Search** 367/125, 129, 367/118, 123

(56) **References Cited**

U.S. PATENT DOCUMENTS

4,688,045 A	*	8/1987	Knudsen	367/123
5,465,302 A		11/1995	Lazzari et al.	381/92
5,778,082 A		7/1998	Chu et al.	381/92

OTHER PUBLICATIONS

Harvey F. Silverman et al., "The Huge Microphone Array (HMA)," May 1996 (published at http://www.lems.brown.edu/array/papers/).

Douglas E. Sturim et al., "Tracking Multiple Talkers Using Microphone Array Measurements," (Apr. 1997) (published at http://www.lems.brown.edu/array/papers/).

Michael S. Brandstein et al., "A Robust Method for Speech Signal Time–Delay Estimation in Reverberant Rooms," (Apr. 1997) (published at http://www.lems.brown.edu/array/papers/).

Michael S. Brandstein et al., "A Closed–Form Location Estimator for Use with Room Environment Microphone Arrays," (Jan. 1997) (published at http://www.lems.brown.edu/array/papers/).

Michael S. Brandstein et al., "A Closed–Form Method for Finding Source Locations From Microphone–Array time–Delay Estimates," (Jan. 1997) (published at http://www.lems.brown.edu/array/papers/).

John E. Adcock, "Optimal Filtering And Speech Recognition With Microphone Arrays," Doctotal Thesis, Brown University (May 2001) (published at http://www.lems.brown.edu/array/papers/).

Michael S. Brandstein et al., "Microphone Array Localization Error Estimation with Application to Sensor Placement" (Jun. 1996) (published at http://www.lems.brown.edu/array/papers/).

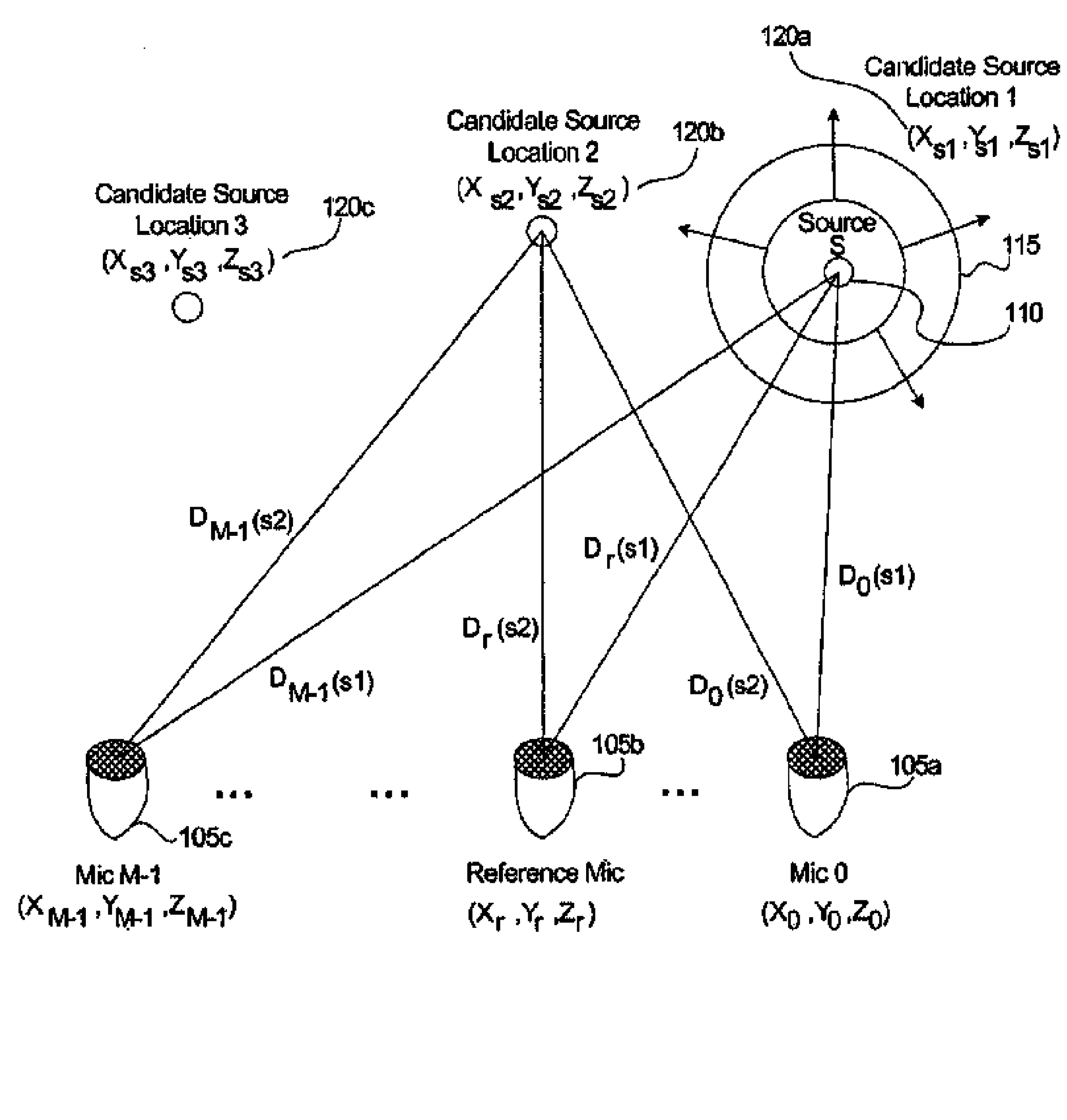
(Continued)

Primary Examiner—Daniel Pihulic
(74) *Attorney, Agent, or Firm*—Wong, Cabello, Lutsch, Rutherford & Brucculeri, LLP

(57) **ABSTRACT**

In accordance with the present invention, a system and method for computing a location of an acoustic source is disclosed. The method includes steps of processing a plurality of microphone signals in frequency space to search a plurality of candidate acoustic source locations for a maximum normalized signal energy. The method uses phase-delay look-up tables to efficiently determine phase delays for a given frequency bin number k based upon a candidate source location and a microphone location, thereby reducing system memory requirements. Furthermore, the method compares a maximum signal energy for each frequency bin number k with a threshold energy $E_t(k)$ to improve accuracy in locating the acoustic source.

22 Claims, 7 Drawing Sheets



OTHER PUBLICATIONS

J. Adcock et al., "Practical Issues in the Use of a Frequency–Domain Delay Estimator for Microphone–Array Applications" (Nov. 1994) (published at http://www.lems.brown.edu/array/papers/).

M.S. Brandstein et al., "A Practical Time–Delay Estimator for Localizing Speech Sources with a Microphone Array" (Sep. 1995) (published at http://www.lems.brown.edu/array/papers/).

M.S. Brandstein, Abstract of"A Framework for Speech Source Localization Using Sensor Arrays," Doctoral Thesis, Brown University (May 1995) (published at http://www.lems.brown.edu/array/papers/).

Paul C. Meuse et al., "Characterization of Talker Radiation Pattern Using a Microphone–Array" (May 1994) (published at http://www.lems.brown.edu/array/papers/).

M.S. Brandstein et al., "A Localization–Error Based Method for Microphone–Array Design" (May 1996) (published at http://www.lems.brown.edu/array/papers/).

John E. Adcock, "Microphone–Array Speech Recognition via Incremental MAP Training," (May 1996) (published at http://www.lems.brown.edu/array/papers/).

Michael Brandstein & Darren Ward (eds.), "Microphone Arrays: Signal Processing Techniques and Applications," pp. 157–201 (Springer, 2001).

Various materials regarding Aethra's Vega Star Gold product, downloaded from www.if2000.de/daten/ae/vi/vegastargolden.pdf (Jun. 2003) and www.aethra.com/eng/productsservices/videocommunication/vegastargold.asp.

"Intelligent Working Spaces," Chapter 5 (2002) (downloaded from www.itc.it/abstracts2002/chapter5.pdf).

Various materials regarding Vtel's "SmartTrak" product (1998), downloaded from www.vtel.com/support/catchall/smrtraks.htm; www.vtel.com/support/catchall/smarttra.htm; www.vtel.com/support/catchall/strakgde.htm; and www.vtel.com/support/catchall/strakins.htm.

Information regarding PictureTel's Limelight product (1998), downloaded from www.polycom.com/common/pw_item_show_doc/0,1449,538,00.pdf.

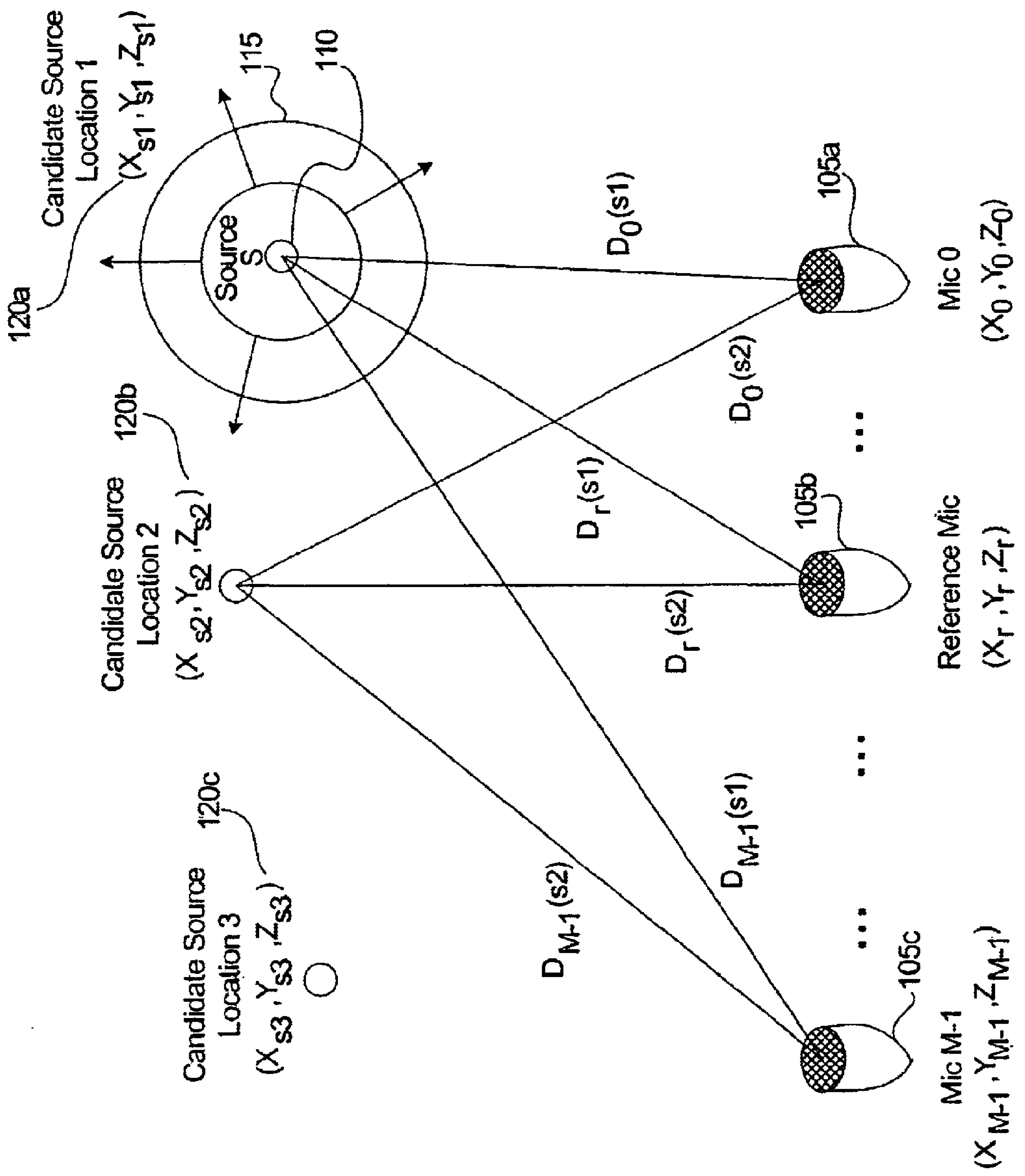
PictureTel's "Concorde 4500 and System 4000EX/ZX Troubleshooting Guide" (1997), downloaded from www.polycom.com/common/pw_item_show_doc/0,1449,444,00.pdf.

PictureTel's "Concorde 4500 Software Version 6.11 Release Bulletin" (1996), downloaded from www.polycom.com/common/pw_item_show_doc/0,1449,438,00.pdf.

PictureTel's "Venue2000 User's Notebook" (1999), downloaded from www.polycom.com/common/pw_item_show_doc/0,1449,675,00.pdf.

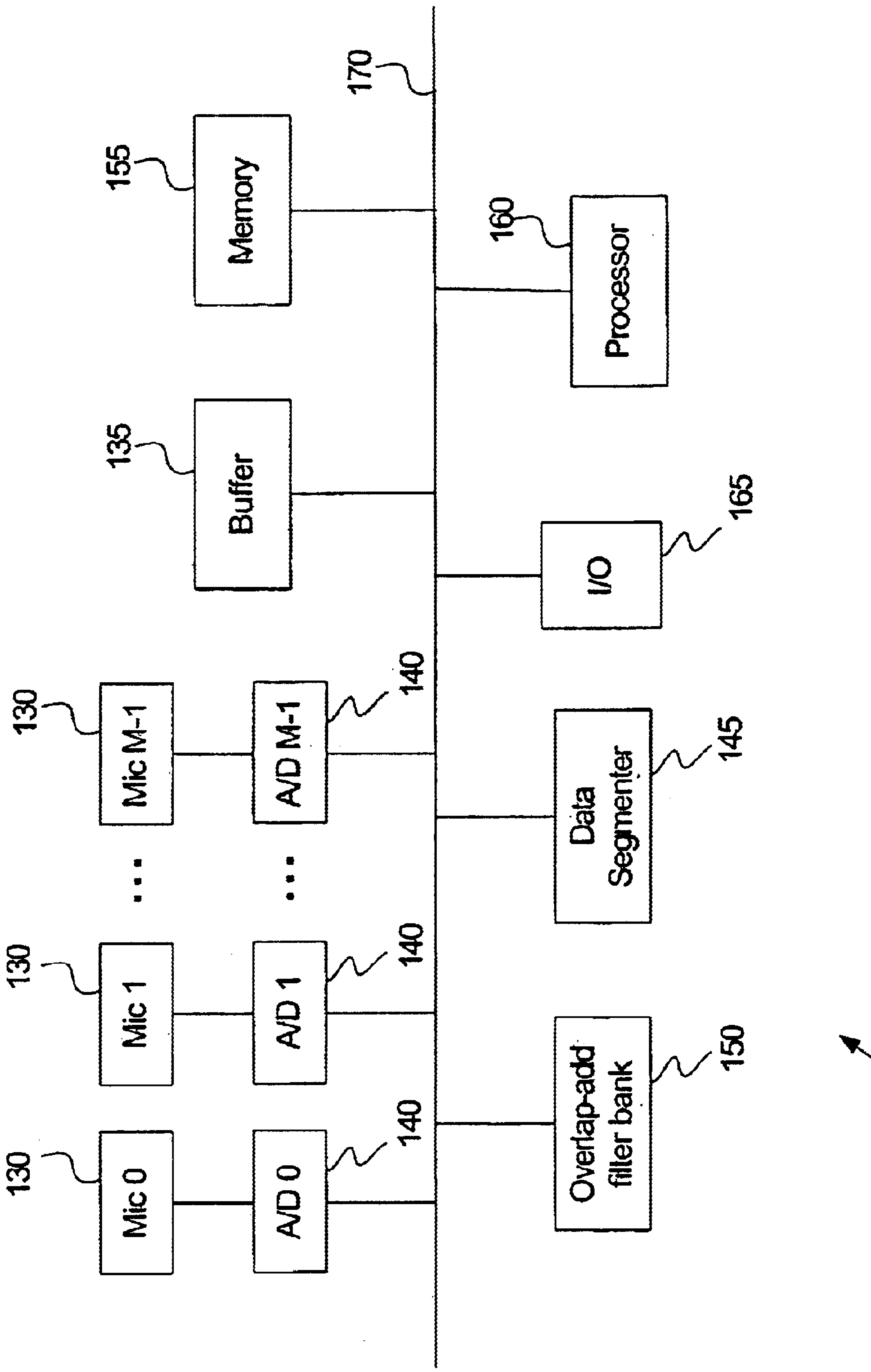
Information regarding PictureTel's 760XL Videoconferencing System (2000), downloaded from www.polycom.com/common/pw_item_show_doc/0,1449,427,00.pdf.

* cited by examiner



120a
Candidate Source Location 1
(X_{s1}, Y_{s1}, Z_{s1})
115
110
Source S
120b
Candidate Source Location 2
(X_{s2}, Y_{s2}, Z_{s2})
120c
Candidate Source Location 3
(X_{s3}, Y_{s3}, Z_{s3})
$D_0(s1)$
$D_0(s2)$
$D_r(s1)$
$D_r(s2)$
$D_{M-1}(s1)$
$D_{M-1}(s2)$
105a
Mic 0
(X_0, Y_0, Z_0)
105b
Reference Mic
(X_r, Y_r, Z_r)
105c
Mic M-1
$(X_{M-1}, Y_{M-1}, Z_{M-1})$

FIG. 1A



170
155
Memory
160
Processor
135
Buffer
165
I/O
130
Mic M-1
A/D M-1
140
145
Data Segmenter
130
Mic 1
A/D 1
140
130
Mic 0
A/D 0
140
150
Overlap-add filter bank
125

FIG. 1B

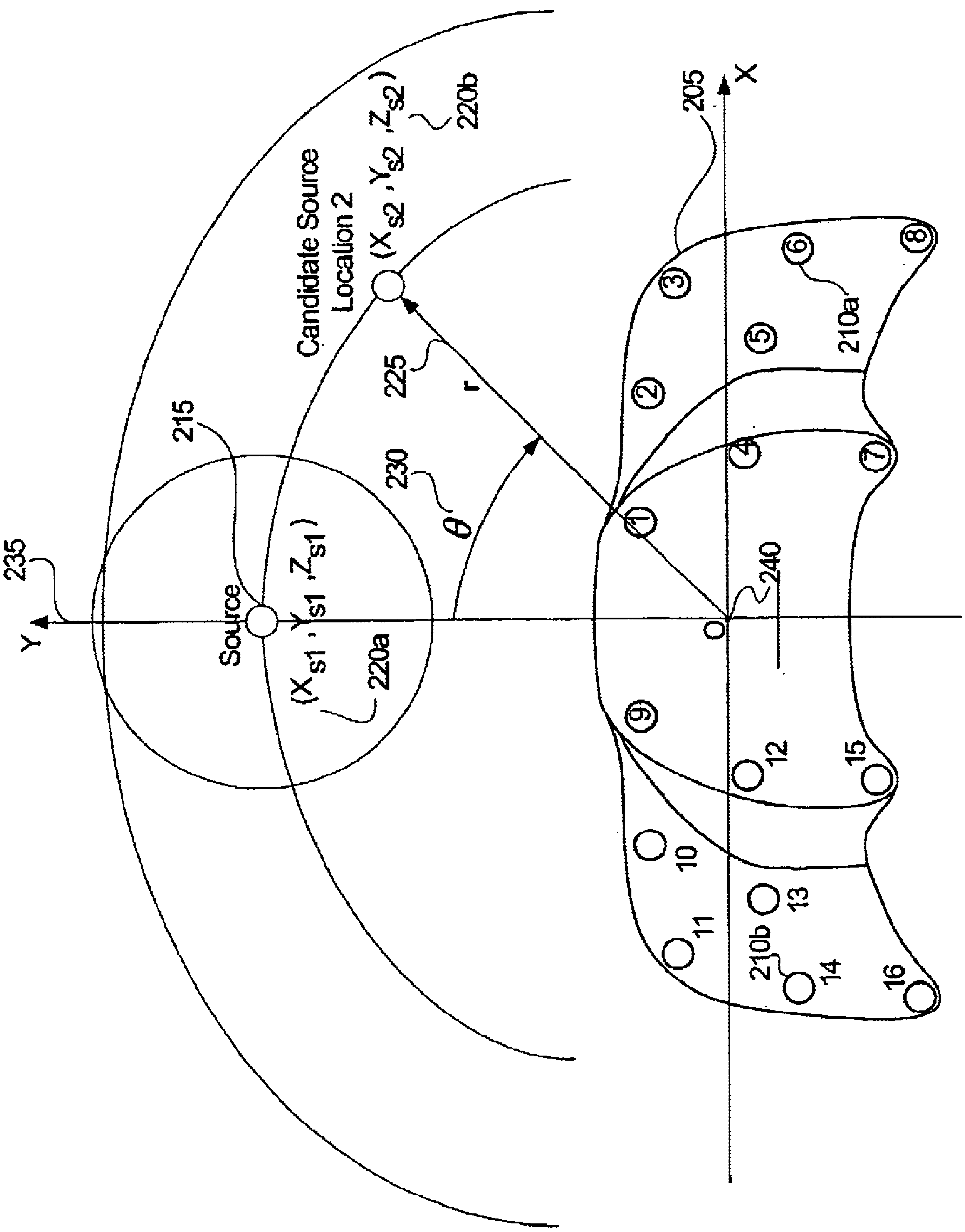

Y
235
Source
(X s1 , Y s1 , Z s1)
220a
215
230
θ
225
r
Candidate Source Location 2
(X s2 , Y s2 , Z s2)
220b
205
X
O
240
1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16
210a
210b

FIG. 2A

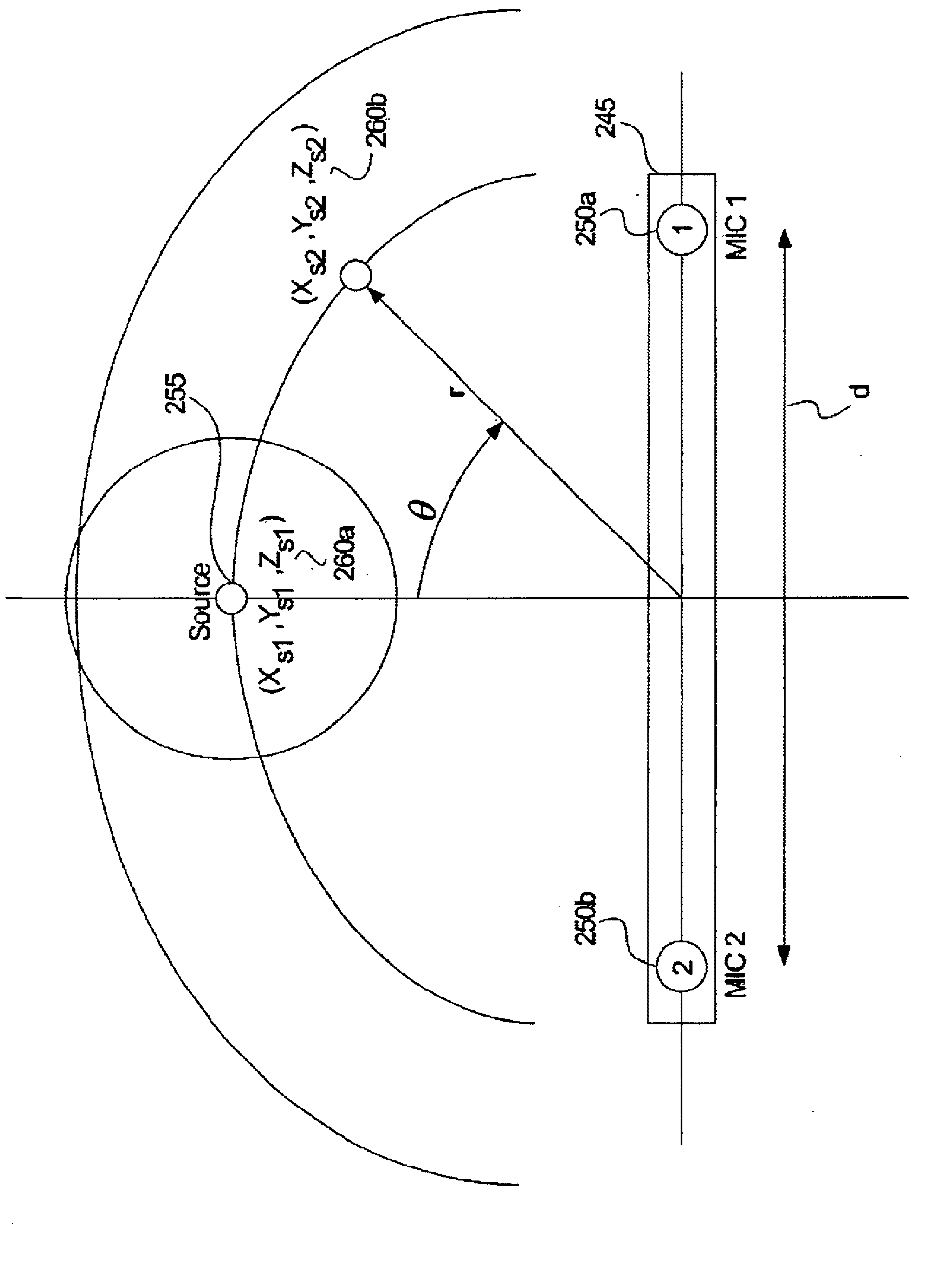
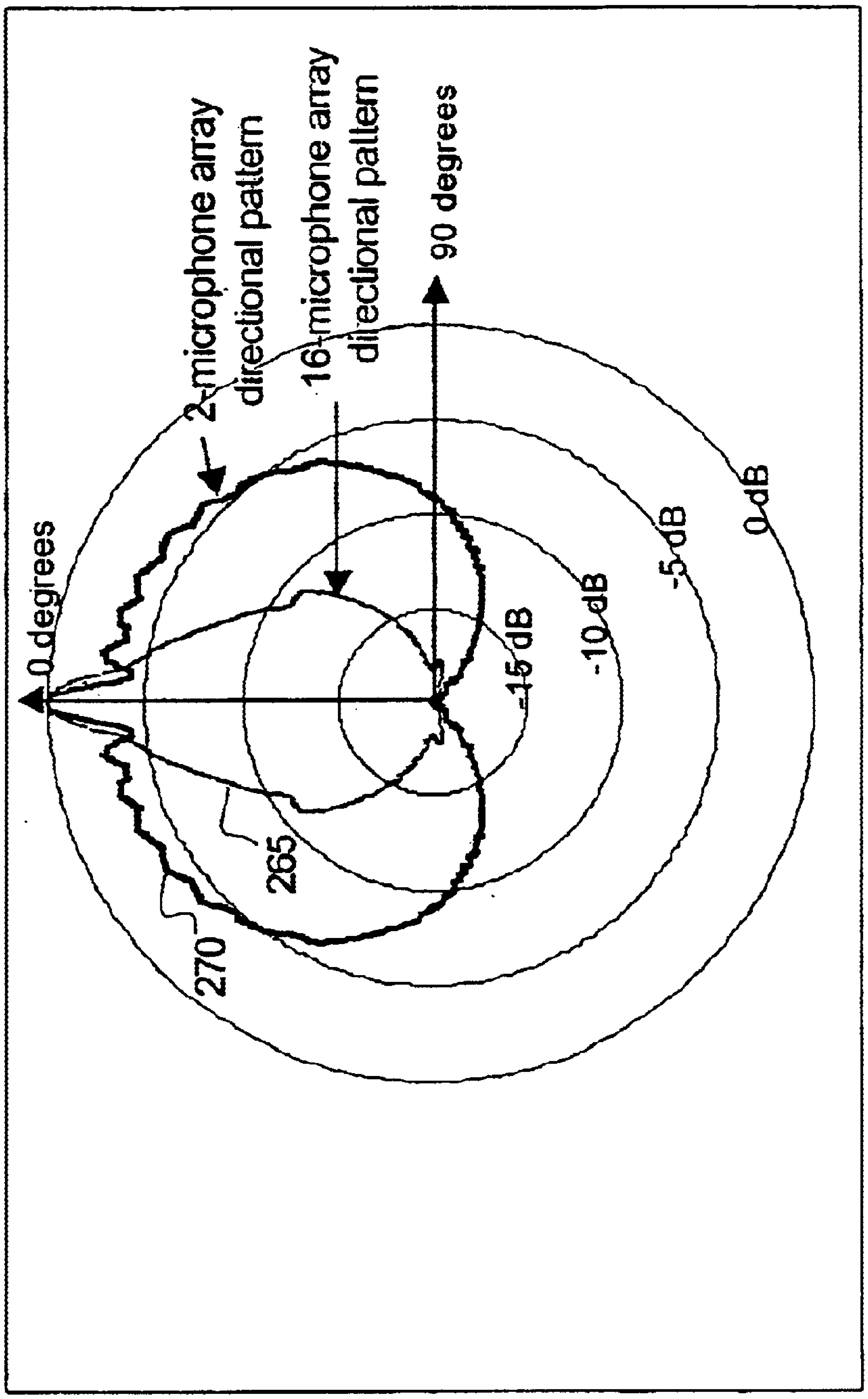

255
Source
(Xs1, Ys1, Zs1)
260a
θ
r
(Xs2, Ys2, Zs2)
260b
245
250a
1
MIC 1
250b
2
MIC 2
d

FIG. 2B



2-microphone array
directional pattern
16-microphone array
directional pattern
90 degrees
0 degrees
0 dB
-5 dB
-10 dB
-15 dB
265
270

FIG. 2C

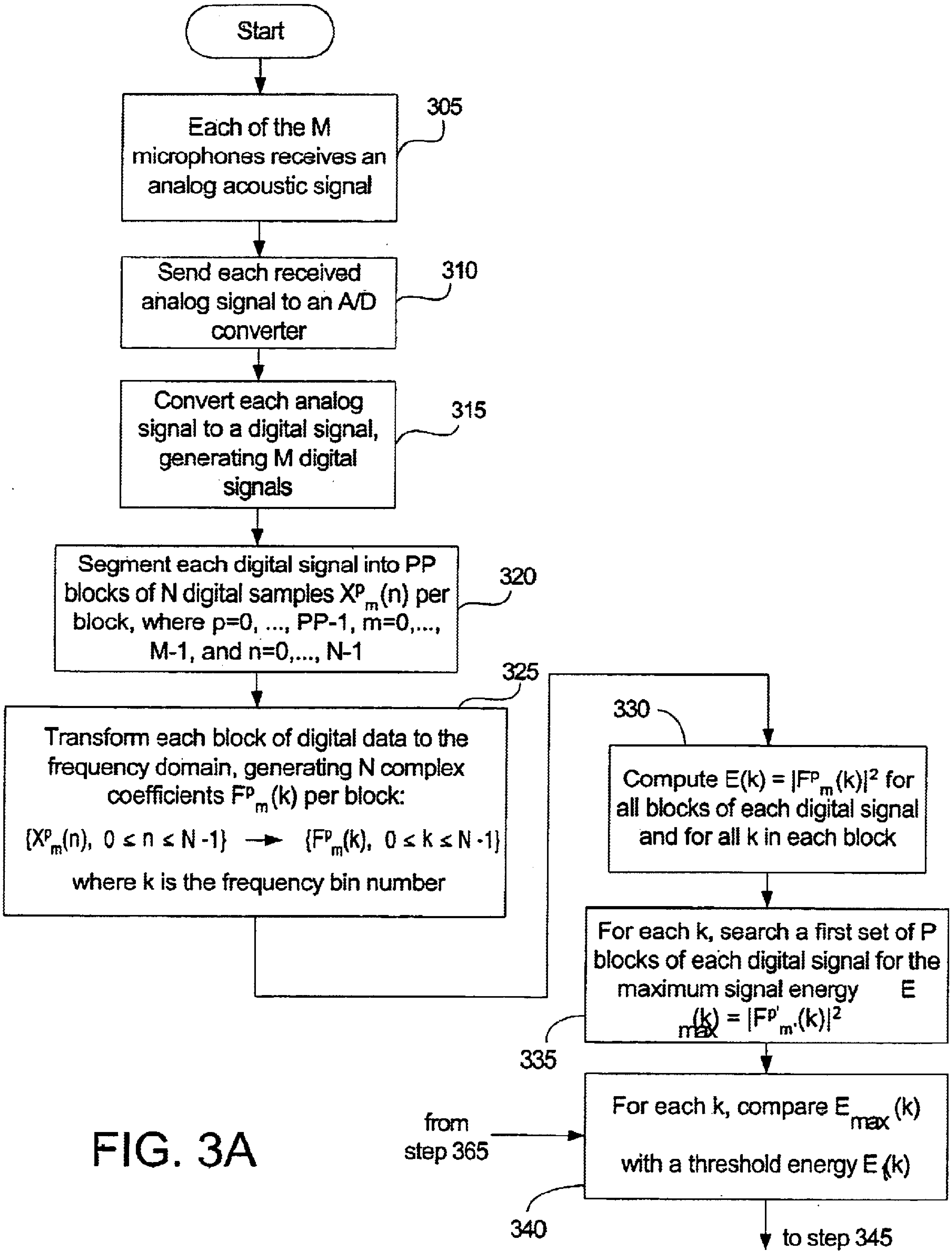

Start
Each of the M microphones receives an analog acoustic signal
305
Send each received analog signal to an A/D converter
310
Convert each analog signal to a digital signal, generating M digital signals
315
Segment each digital signal into PP blocks of N digital samples $X^p_m(n)$ per block, where p=0, ..., PP-1, m=0,..., M-1, and n=0,..., N-1
320
Transform each block of digital data to the frequency domain, generating N complex coefficients $F^p_m(k)$ per block:
$\{X^p_m(n),\ 0 \le n \le N-1\} \rightarrow \{F^p_m(k),\ 0 \le k \le N-1\}$
where k is the frequency bin number
325
330
Compute $E(k) = |F^p_m(k)|^2$ for all blocks of each digital signal and for all k in each block
For each k, search a first set of P blocks of each digital signal for the maximum signal energy $E_{max}(k) = |F^{p'}_{m'}(k)|^2$
335
from step 365
For each k, compare $E_{max}(k)$ with a threshold energy $E_t(k)$
340
to step 345

FIG. 3A

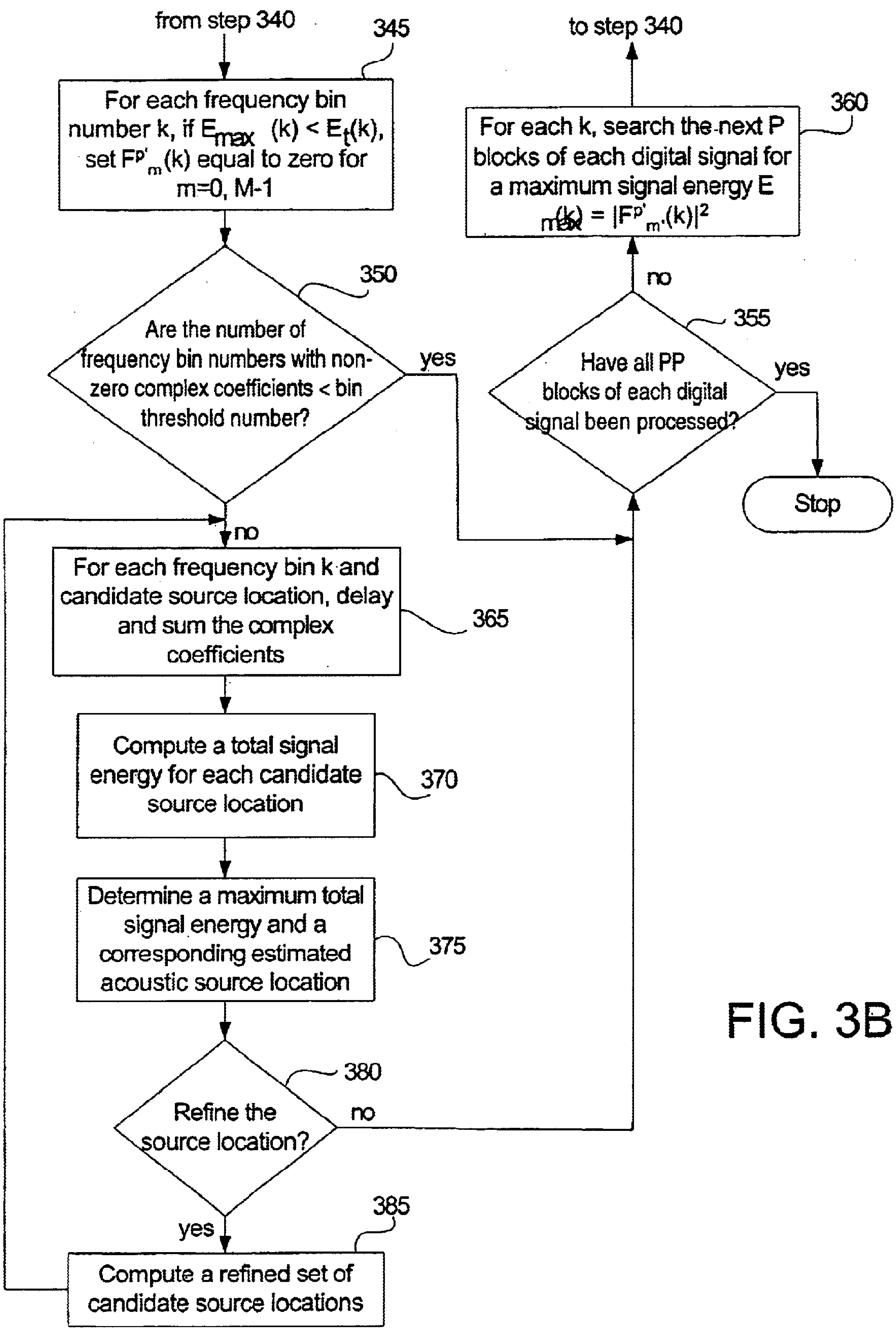

from step 340
345
For each frequency bin number k, if E max (k) < Et(k), set FP'm(k) equal to zero for m=0, M-1
350
Are the number of frequency bin numbers with non-zero complex coefficients < bin threshold number?
yes
no
For each frequency bin k and candidate source location, delay and sum the complex coefficients
365
Compute a total signal energy for each candidate source location
370
Determine a maximum total signal energy and a corresponding estimated acoustic source location
375
Refine the source location?
380
no
yes
385
Compute a refined set of candidate source locations
to step 340
360
For each k, search the next P blocks of each digital signal for a maximum signal energy E max(k) = |FP'm'(k)|2
no
355
Have all PP blocks of each digital signal been processed?
yes
Stop

FIG. 3B

SYSTEM AND METHOD FOR COMPUTING A LOCATION OF AN ACOUSTIC SOURCE

CROSS REFERENCE TO RELATED APPLICATION

This application claims the benefit of Provisional Patent Application Ser. No. 60/372,888, filed Apr. 15, 2002, entitled "Videoconferencing System with Horizontal and Vertical Microphone Arrays for Enhanced Source Locating and Camera Tracking," which is incorporated herein by reference. This application is related to U.S. application Ser. No. 10/414,420 entitled "Videoconferencing System with Horizontal and Vertical Microphone Arrays", by Peter Chu, Michael Kenoyer, and Richard Washington, filed concurrently herewith, which is incorporated herein by reference.

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates generally to signal processing and more particularly to a method for computing a location of an acoustic source.

2. Description of the Background Art

Spatial localization of people talking in a room is important in many applications, such as surveillance and videoconferencing applications. In a videoconferencing application, a camera uses spatial localization data to track an acoustic source. Typically, a videoconferencing system localizes the acoustic source by applying cross correlation techniques to signals detected by a pair of microphones. The cross correlation techniques involve finding the crosscorrelation between the time domain signals of a pair of microphones. The shift in time which corresponds to the peak of the cross correlation corresponds to the difference in time of arrival of the acoustic source to the two microphones. Knowledge of the difference in time of arrival infers that the source is located in a geometric plane in space. By using three pairs of microphones, one can locate the source by finding the intersection of the three planes.

However, the 2-microphone cross correlation techniques of the prior art provide slow, inaccurate, and unreliable spatial localization of acoustic sources, particularly acoustic sources located in noisy, reverberant environments. A primary reason for the poor performance of the two-microphone cross correlation techniques for estimating an acoustic source location is poor sidelobe attenuation of a directional pattern formed by delaying and summing the two microphone signals. For example, an acoustic source located in a reverberant environment, such as a room, generates acoustic signals which are reflected from walls and furniture. Reflected signals interfere with the acoustic signals that are directly propagated from the acoustic source to the microphones. For a 2-microphone array, the direct and reflected acoustic signals received by the microphones may increase sidelobe magnitude of the 2-microphone directional pattern, and may produce an erroneous acoustic source location. The poor sidelobe attenuation of the 2-microphone directional pattern is further discussed below in conjunction with FIG. 2C.

It would be advantageous to designers of surveillance and videoconferencing applications to implement an efficient and accurate method for spatial localization of acoustic sources, particularly acoustic sources located in noisy and reverberant environments.

SUMMARY OF THE INVENTION

In accordance with the present invention, a system and method for computing a location of an acoustic source is

disclosed. In one embodiment of the invention, the present system includes a plurality of microphones for receiving acoustic signals generated by the acoustic source, at least one A/D converter for digitizing the acoustic signals received by the plurality of microphones, a data segmenter for segmenting each digitized signal into a plurality of blocks, an overlap-add filter bank for generating a plurality of transformed blocks by performing a Fast Fourier Transform (FFT) on each block, a memory configured to store phase-delay look-up tables, and a processor for computing the location of the acoustic source by processing the transformed blocks of each acoustic signal received by each microphone according to candidate source locations using the phase-delay look-up tables.

In one embodiment of the invention, the method for computing the location of the acoustic source includes receiving a plurality of M analog signals from a plurality of M microphones, digitizing each received analog signal, segmenting each digitized signal into a plurality of blocks, performing a discrete Fast Fourier Transform (FFT) on each block to generate N complex coefficients $F^p_m(k)$ per block, searching P blocks of each digitized signal for a maximum signal energy and identifying a block p' containing the maximum signal energy for each frequency bin number k, comparing the maximum signal energy with a threshold energy $E_t(k)$ and setting the complex coefficients in the P blocks of each digitized signal equal to zero when the maximum signal energy is less than the threshold energy for each frequency bin number k, determining phase delays using three look-up tables, multiplying each complex coefficient by an appropriate phase delay and summing the phase-delayed complex coefficients over the M microphones for each candidate source location and for each frequency bin number k, computing a normalized total signal energy for each candidate source location, and finally determining the location of the acoustic source based upon the normalized total signal energy for each candidate source location.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1A is a diagram illustrating nomenclature and elements of an exemplary microphone/source configuration, according to one embodiment of the present invention;

FIG. 1B is an exemplary block diagram of a system for locating an acoustic source, according to one embodiment of the present invention;

FIG. 2A is an exemplary embodiment a 16-microphone array, according to the present invention;

FIG. 2B is an exemplary embodiment of a 2-microphone array, according to the present invention;

FIG. 2C is an exemplary polar plot of a total signal energy received by the 16-microphone array of FIG. 2A and a total signal energy received by the 2-microphone array of FIG. 2B;

FIG. 3A is flowchart of exemplary method steps for estimating an acoustic source location, according to the present invention; and

FIG. 3B is a continuation of the FIG. 3A flowchart of exemplary method steps for estimating the acoustic source location, according to the present invention.

DETAILED DESCRIPTION OF THE DRAWINGS

FIG. 1A is a diagram illustrating nomenclature and elements of an exemplary microphone/source configuration, according to one embodiment of the present invention. Elements of FIG. 1A include a plurality of microphones

105a–c (also referred to as a microphone array) configured to receive acoustic signals from a source S **110**, a candidate source location **1** (X_{s1}, Y_{s1}, Z_{s1}) **120**a, a candidate source location **2** (X_{s2}, Y_{s2}, Z_{s2}) **120**b, and a candidate source location **3** (X_{s3}, Y_{s3}, Z_{s3}) **120**c. An integer microphone index m is used to label each microphone, where $0 \leqq m \leqq M-1$, M is a total number of microphones, and M is any integer greater than 1. For ease of illustration, the FIG. **1**A embodiment shows three microphones **105**a–c (i.e., M=3), although the present invention typically uses a larger number of microphones (i.e., M>3). The plurality of microphones **105**a–c include a microphone **0** **105**a located at (X_0, Y_0, Z_0), a reference microphone **105**b located at (X_r, Y_r, Z_r), and a microphone M-**1** **105**c located at (X_{M-1}, Y_{M-1}, Z_{M-1}). According to the present invention, one microphone of the plurality of microphones **105**a–c is designated as the reference microphone **105**b, where any one of the plurality of microphones **105**a–c may be designated as a reference microphone.

The source S **110** is any acoustic source for generating acoustic signals. For example, the source S **110** may be a person, an electrical apparatus, or a mechanical apparatus for generating acoustic signals. The acoustic signals generated by the source S **110** propagate away from the source S **110**. Concentric circles **115** centered about the source S **110** are projections of spherical wave fronts generated by the source S **110** onto a two-dimensional plane of FIG. **1**.

For the purposes of the following discussion, the source S **110** is located at the candidate source location **1** (X_{s1}, Y_{s1}, Z_{s1}) **120**a. However, the scope of the present invention covers the source S **110** located at any one of the plurality of candidate source locations **120**a–c, or at a location that does not coincide with any of the plurality of candidate source locations **120**a–c.

The present invention computes a total signal energy received from each of the plurality of candidate source locations **120**a–c by appropriately delaying the microphone signals with respect to a signal received by the reference microphone **105**b, and then summing the delayed signals. The present invention may be implemented as application software, hardware, or application software/hardware (firmware). Although FIG. **1**A illustrates three candidate source locations (**120**a, **120**b, **120**c), the present invention includes any number of candidate source locations. In addition, although a location of the source S **110** may or may not correspond to one of the candidate source locations **120**a–c, the present invention estimates the location of the source S **110** at one of the plurality of candidate source locations **120**a–c, based upon the total signal energy computed for each candidate source location.

FIG. **1**A also shows distances measured from each microphone **105**a–c to the candidate source locations **120**a and **120**b. For example, D_0(s**1**) is a distance between the microphone **0** **105**a and the candidate source location **1** **120**a, D_0(s**2**) is a distance between the microphone **0** **105**a and the candidate source location **2** **120**b, D_r(s**1**) is a distance between the reference microphone **105**b and the candidate source location **1** **120**a, D_r(s**2**) is a distance between the reference microphone **105**b and the candidate source location **2** **120**b, D_{M-1}(s**1**) is a distance between the microphone **105**c and the candidate source location **1** **120**a, and D_{M-1}(s**2**) is a distance between the microphone **105**c and the candidate source location **2** **120**b. Although FIG. **1**A shows the microphones **105**a–c, the candidate source locations **120**a–c, and the source S **110** constrained to lie in the two-dimensional plane of FIG. **1**A, the scope of the present invention covers any two-dimensional or three-dimensional configuration of the microphones **105**a–c, the candidate source locations **120**a–c, and the source S **110**.

FIG. **1**B is an exemplary block diagram of a system **125** for locating an acoustic source, such as the acoustic source S **110** (FIG. **1**A), according to one embodiment of the present invention. Elements of the system **125** include M microphones **130** for receiving M acoustic signals generated by the acoustic source S **110**, at least one analog/digital (A/D) converter **140** for converting the M received acoustic signals to M digital signals, a buffer **135** for storing the M digital signals, a data segmenter **145** for segmenting each digital signal into blocks of data, an overlap-add filter bank **150** for performing a Fast Fourier Transform (FFT) on each block of data for each digital signal, a memory **155** for storing acoustic source location software, look-up tables, candidate source locations, and initialization parameters/constants associated with determining the acoustic source location, a processor **160** for executing the acoustic source location software and for signal processing, an input/output (I/O) port **165** for receiving/sending data from/to external devices (not shown), and a bus **170** for electrically coupling the elements of the system **125**.

According to the present invention, one method of locating the source S **110** is using a maximum likelihood estimate. Using the maximum likelihood estimate, the source S **110** is hypothesized to be located at a plurality of possible candidate locations, such as the candidate source locations **120**a, **120**b, and **120**c (FIG. **1**A). The maximum likelihood estimate may be implemented with the acoustic source location software stored in the memory **155** and executed by the processor **160**, or acoustic source location firmware. In one embodiment of the method for computing an acoustic source location, the analog-to-digital (A/D) converter **140** digitizes each signal received by each microphone **130**. Then, the data segmenter **145** segments each digitized signal into blocks of data. Next, the overlap-add filter bank **150** performs a discrete Fast Fourier Transform (FFT) on each block of data. For example, if each signal received by each microphone **130** is digitized and segmented into blocks of data, where each block of data includes N=640 digitized time samples, then each block of data sampled in time is mapped to a block of data sampled in frequency, where each data sampled in frequency is a complex number (also called a complex coefficient), and each block of data sampled in frequency includes N=640 discrete frequency samples. Each complex coefficient is associated with a frequency bin number k, where $0 \leqq k \leqq N-1$ and k is an integer.

Then, for each candidate source location **120**a–c and for each frequency bin number k, each complex coefficient associated with each microphone's signal is multiplied by an appropriate phase delay, the complex coefficients are summed over all the microphone signals, and a signal energy is computed. A whitening filter is then used to normalize the signal energy for each frequency bin number k, and the normalized signal energies are summed over the N frequency bin numbers for each candidate source location **120**a–c to give a total signal energy for each candidate source location **120**a–c. The method then determines the candidate source location **120**a–c associated with a maximum total signal energy and assigns this candidate source location as an estimated location of the source S **110**. A computationally efficient method of implementing the maximum likelihood estimate for estimating an acoustic source location will be discussed further below in conjunction with FIGS. **3**A–**3**B.

FIG. **2**A illustrates one embodiment of a 16-microphone array **205** for receiving acoustic signals to be processed by acoustic source location software or firmware, according to the present invention. The 16-microphone array **205** includes an arrangement of 16 microphones labeled **1–16** configured to receive acoustic signals from an acoustic source **215**. In the present embodiment, a distance between a microphone **6 210***a* and a microphone **14 210***b* is 21.5 inches. Thus, the 16-microphone array **205** spans 21.5 inches. The acoustic source **215** is located at a candidate source location **1** (X_{s1},Y_{s1}, Z_{s1}) **220***a*. The acoustic source location software or firmware processes the signals received by the 16 microphones according to candidate source locations. For simplicity of illustration, the FIG. **2**A embodiment shows only candidate source locations **220***a* and **220***b*, but the scope of the present invention includes any number of candidate source locations. Although FIG. **2**A illustrates a specific spatial distribution of the 16 microphones as embodied in the 16-microphone array **205**, the present invention covers any number of microphones distributed in any two-dimensional or three-dimensional spatial configuration. In the FIG. **2**A embodiment of the present invention, each candidate source location may be expressed in polar coordinates. For example, the candidate source location **2 220***b* has polar coordinates (r,θ), where r is a magnitude of a vector r 225 and θ **230** is an angle subtended by the vector r **225** and a positive y-axis **235**. The vector r **225** is a vector drawn from an origin O **240** of the 16-microphone array **205** to the candidate source location **2 220***b*, but the vector r **225** may be drawn to the candidate source location **1 220***a*, or to any candidate source location of a plurality of candidate source locations (not shown).

FIG. **2**B is one embodiment of a 2-microphone array **245** for receiving acoustic signals to be processed by acoustic source location software or firmware, according to the present invention. The microphone array **245** includes a microphone **1 250***a* and a microphone **2 250***b* separated by a distance d, an acoustic source **255** located at a candidate source location **1** (X_{s1}, Y_{s1}, Z_{s1}) **260***a*, and a candidate source location **2** (X_{s2}, Y_{s2}, Z_{s2}) **260***b*. Although the FIG. **2**B embodiment of the present invention illustrates two candidate source locations (**260***a* and **260***b*), the present invention covers any number of candidate source locations. In one embodiment of the present invention d=21.5 inches, although in other embodiments of the invention the microphone **1 250***a* and the microphone **2 250***b* may be separated by any distance d.

FIG. **2**C shows a 16-microphone array polar plot of total signal energy computed by acoustic source location software (i.e., application software), or firmware upon receiving acoustic signals from the acoustic source **215** (FIG. **2**A) and the 16-microphone array **205** (FIG. **2**A), and a 2-microphone array polar plot of total signal energy computed by the application software or firmware upon receiving acoustic signals from the acoustic source **255** (FIG. **2**B) and the 2-microphone array **245** (FIG. **2**B). The polar plots are also referred to as directional patterns. As illustrated in FIG. **2**A and FIG. **2**B, the source **215** is located at θ=0 degrees and the source **255** is located at θ=0 degrees, respectively. In generating the FIG. **2**C embodiments of the 16-microphone array polar plot and the 2-microphone array polar plot, the source **215** and the source **255** are identical white noise sources spanning a frequency range of 250 Hz to 5 kHz. FIG. **2**C illustrates that a magnitude of a sidelobe **265** of the 16-microphone array directional pattern is smaller than a magnitude of a sidelobe **270** of the 2-microphone array directional pattern, where sidelobe magnitude is measured in decibels (dB).

Spurious acoustic signals may be generated by reflections of acoustic source signals from walls and furnishings of a room. These spurious signals may interfere with the 2-microphone array directional pattern and the 16-microphone array directional pattern computed by the application software as illustrated in FIG. **2**C. However, since the sidelobe **270** of the 2-microphone array directional pattern is greater in magnitude than the sidelobe **265** of the 16-microphone array directional pattern, the spurious signals may cause a greater uncertainty in an estimated location of the acoustic source **255** using the 2-microphone array **245**. For example, the spurious signals may increase the magnitude of the sidelobe **270** of the 2-microphone array directional pattern to zero dB, thus generating an uncertainty in the estimated location of the acoustic source **255**. More specifically, poor sidelobe attenuation of the 2-microphone array directional pattern may allow spurious signals to interfere with accurately estimating an acoustic source location. Thus, application software or firmware for processing acoustic signals received by the 16-microphone array **205** is the preferred embodiment of the present invention, however, the scope of the present invention covers application software or firmware for processing acoustic signals received by microphone arrays having any number of microphones distributed in any two-dimensional or three-dimensional configuration.

Since the scope of the present invention includes processing acoustic signals received by a plurality of microphones to search thousands of candidate source locations, a straightforward implementation of the maximum likelihood estimate method is computationally intense. Accordingly, the present invention uses a plurality of microphones and a computationally efficient implementation of the maximum likelihood estimate method to compute a location of an acoustic source in an accurate manner.

FIG. **3**A is flowchart of exemplary method steps for estimating an acoustic source location, according to the present invention. In step **305**, each of the M microphones (FIG. **1**A, FIG. **1**B, FIG. **2**A, or FIG. **2**B) receives an analog acoustic signal $x_m(t)$, where m is an integer microphone index which identifies each of the microphones, and $0 \leqq m \leqq M-1$. In step **310**, each microphone sends each received analog signal to an associated A/D converter **140** (FIG. **1**B). For example, in one embodiment of the invention, each analog signal received by the microphone **130** (FIG. **1**B) is sent to each analog signal's associated A/D converter **140** so that in step **315** each associated A/D converter **140** converts each analog signal to a digital signal, generating M digital signals. For example, in one embodiment of the invention, each associated A/D converter **140** samples each analog signal at a sampling rate of f_s=32 kHz. The M digital signals are stored in the buffer **135** (FIG. **1**B).

In step **320**, a data segmenter **145** (FIG. **1**B) segments each digital signal into PP blocks of N digital samples $X^p{}_m(n)$ per block, where PP is an integer, p is an integer block index which identifies a block number ($0 \leqq p \leqq PP-1$), n is an integer sample index which identifies a sample number ($0 \leqq n \leqq N-1$), and m is the integer microphone index which identifies a microphone ($0 \leqq m \leqq M-1$). In one embodiment of the invention, each block is of time length T=0.02 s, and each block comprises N=640 digital samples. However, the scope of the invention includes any time length T, any sampling rate f_s, and any number of samples per block N.

In step **325**, an overlap-add filter bank **150** (FIG. **1**B) performs a discrete Fast Fourier Transform (FFT) on each block of digital samples, (also referred to as digital data), to generate N complex coefficients per block, where each complex coefficient is a function of a discrete frequency identified by a frequency bin number k. More specifically, a set of N digital samples per block is mapped to a set of N complex coefficients per block: $\{X^p_m(n), 0 \leqq n \leqq N-1\} \rightarrow \{F^p_m(k), 0 \leqq k \leqq N-1\}$. The N complex coefficients $F^p_m(k)$ are complex numbers with real and imaginary components.

In step **330**, the method computes a signal energy $E(k)=|F^p_m(k)|^2$ for each complex coefficient ($0 \leqq p \leqq PP-1$ and $0 \leqq m \leqq M-1$) for each frequency bin number k. More specifically, the method computes M×PP signal energies for each frequency bin number k. In this step and all subsequent steps of the FIG. **3A-3B** embodiment of the present invention, methods for performing various functions and/or signal processing are described. In an exemplary embodiment of steps **330–385**, the methods described are performed by the processor **160** (FIG. **1B**) executing acoustic source location software, and in other embodiments, the methods described are performed by a combination of software and hardware.

In step **335**, the method searches, for each frequency bin number k, the signal energies of a first set of P blocks of each digital signal for a maximum signal energy $E_{max}(k)=|F^{p'}_{m'}(k)|^2$, where p' specifies a block associated with the maximum signal energy and m specifies a microphone associated with the maximum signal energy. In one embodiment of the invention, P=5.

Next, in step **340**, the method compares each $E_{max}(k)$ with a threshold energy $E_t(k)$. In one embodiment of the invention, the threshold energy $E_t(k)$ for each frequency bin number k is a function of background noise energy for the frequency bin number k. For example, the threshold energy $E_t(k)$ may be predefined for each frequency bin number k and stored in the memory **155** (FIG. **1B**), or the method may compute the threshold energy $E_t(k)$ for each frequency bin number k using the M microphone signals to compute background noise energy during periods of silence. A period of silence may occur when conference participants are not speaking to the M microphones, for example.

FIG. **3B** is a continuation of the FIG. **3A** flowchart of exemplary method steps for estimating an acoustic source location, according to the present invention. In step **345**, if $E_{max}(k)<E_t(k)$ for a given k, then the complex coefficients are set equal to zero for all values of m for the block p'. That is, if $E_{max}(k)<E_t(k)$, then $F^{p'}_m(k)=0$ for $0 \leqq m \leqq M-1$. The complex coefficients for a given frequency bin number k are set equal to zero when a maximum signal energy associated with those complex coefficients is below a threshold energy. If these complex coefficients are not set equal to zero, then the method may compute an inaccurate acoustic source location due to excessive noise in the acoustic signals. However, as will be seen further below in conjunction with step **365**, if these complex coefficients are set equal to zero, excessive signal noise associated with frequency bin number k is eliminated in the computation of an acoustic source location.

In step **350**, the method determines if the number of frequency bin numbers with non-zero complex coefficients is less than a bin threshold number. The bin threshold number may be a predefined number stored in the memory **155** (FIG. **1B**). The bin threshold number is defined as a minimum number of frequency bin numbers with non-zero complex coefficients that the method requires to compute an acoustic source location. If, in step **350**, the method determines that the number of frequency bin numbers with non-zero complex coefficients is less than the bin threshold number, then the total signal strength is too weak to accurately compute an acoustic source location, steps **365–385** are bypassed, and in step **355**, the method determines if all PP blocks of each digital signal have been processed. If, in step **355**, the method determines that all PP blocks of each digital signal have been processed, then the method ends. If, in step **355**, the method determines that all PP blocks of each digital signal have not been processed, then in step **360**, the method searches, for each frequency bin number k, a next set of P blocks of each digital signal for a maximum signal energy $E_{max}(k)=|F^{p'}_{m'}(k)|^2$. Then, steps **340–350** are repeated.

If, in step **350**, the method determines that the number of frequency bin numbers with non-zero complex coefficients is greater than or equal to the bin threshold number, then in step **365**, the complex coefficients are phase-delayed and summed over the microphone index m for each frequency bin number k and each candidate source location. Each phase delay θ_m is a function of the frequency bin number k, a candidate source location, and a microphone location (as represented by the microphone index m) with respect to a reference microphone location. For example, for a given frequency bin number k and a candidate source location (x,y,z), a summation over the index m of the phase-delayed complex coefficients is

$$G_{x,y,z}(k) = \sum_{m=0}^{M-1} e^{j\theta m}$$

$F^{p'}_m(k)$, where the complex coefficients $F^{p'}_m(k)$ from block p' are phase-delayed and summed, and where p' is the block associated with the maximum signal energy for the given frequency bin number k.

A phase delay between a microphone m (i.e., a microphone corresponding to the microphone index m), and a reference microphone, such as the reference microphone **105***b* (FIG. **1A**), is $\theta_m=2\pi kb\Delta_m v$, where b is a width of each frequency bin number k in Hertz, v is a constant that is proportional to a reciprocal of the speed of sound (i.e., an acoustic signal speed), and Δ_m is a difference in distance between a location (X_m,Y_m,Z_m) of the microphone m and a candidate source location (x,y,z), and a location (X_r,Y_r,Z_r) of the reference microphone and the candidate source location (x,y,z). For example, $\Delta_m=D_m-D_r$, where $D_m=((x-X_m)^2+(y-Y_m)^2+(z-Z_m)^2)^{1/2}$ is the distance between the candidate source location (x,y,z) and the location (X_m, Y_m,Z_m) of the microphone m, and $D_r=((x-X_r)^2+(y-Y_r)^2+(z-Z_r)^2)^{1/2}$ is the distance between the candidate source location (x,y,z) and the location (X_r,Y_r,Z_r) of the reference microphone. Space surrounding a microphone array, such as the 16-microphone array **205** (FIG. **2A**), may be divided up into a plurality of coarsely or finely separated candidate source locations. For example, in one embodiment of the invention, the candidate source locations may be located along a circle with the microphone array placed at the center of the circle. In this embodiment, a radius of the circle is 10 feet, and 61 candidate source locations are place at three degree increments along the circle, spanning an angle of 180 degrees. Then, for each k and for each candidate source location, the method computes a sum over the index m of the phase-delayed complex coefficients.

In step **370**, the method computes a total signal energy for each candidate source location. The total signal energy is

$$W(x, y, z) = \sum_k [|G_{x,y,z}(k)|^2 / |S(k)|^2],$$

where a total energy $|G_{x,y,z}(k)|^2$ received by the M microphones in the frequency bin number k from the candidate source location (x,y,z) is normalized by a whitening term $|S(k)|^2$. The whitening term is an approximate measure of the signal strength in frequency bin number k. In one embodiment of the present invention, the method computes $|S(k)|^2$ by averaging the signal energy of all the microphone signals for a given k, where

$$|S(k)|^2 = \sum_m \left|F_m^{p'}(k)\right|^2.$$

Normalization of the total energy $|G_{x,y,z}(k)|^2$ of frequency bin number k by the whitening term $|S(k)|^2$ allows all frequency components of an acoustic source to contribute to the computation of a location of the acoustic source.

Typically, the total signal energy W(x,y,z) is computed by a summation over k, where k=0, . . . , N-**1**. However, the scope of the present invention also includes a trimmed frequency summation, where k is summed from a low frequency bin number (k_{low}>0) to a high frequency bin number (k_{high}<N-**1**). By ignoring the very low and the very high frequency components in the summation of the total signal energy, cost to compute a location of the acoustic source is reduced.

In step **375**, the method determines a maximum total signal energy, and thus a candidate source location associated with the maximum total signal energy. The candidate source location associated with the maximum total signal energy is identified as an estimated location of the acoustic source.

In step **380**, if the location of the acoustic source is to be refined, then in step **385**, the method computes a refined set of candidate source locations. For example, in one embodiment of the invention, the computed refined set of candidate source locations are centered about the acoustic source location computed in step **375**. In another embodiment of the invention, the method uses a refined set of candidate source locations stored in the memory **155**. For example, the stored refined set of candidate source locations may be located along six concentric rings in a quarter of a degree increments along each ring, where each concentric ring has a unique radius and each concentric ring spans 180 degrees. In this embodiment of the invention, there are 4326 refined candidate source locations. As discussed further below in conjunction with a more detailed description of step **365**, the stored refined candidate source locations may be incorporated in look-up tables stored in the memory **155**.

Next, steps **365**–**380** are repeated, and a refined acoustic source location is computed. However, if in step **380**, a refinement to the acoustic source location is not desired, then in step **355**, the method determines if all PP blocks of each digital signal have been processed. If all PP blocks of each digital signal have been processed, then the method ends. If, in step **355**, all PP blocks of each digital signal have not been processed, then in step **360** the method searches, for each frequency bin number k, the next set of P blocks of each digital signal for a maximum signal energy $E_{max}(k)=|F^{P'}_{m'}(k)|^2$, and the method continues at step **340**.

Referring back to step **365**, the method phase-delays each complex coefficient by multiplying each complex coefficient with a transcendental function $e^{i\theta_m}=\cos(\theta_{m'})+i\cdot\sin(\theta_m)$. It is costly and inefficient to compute the transcendental function at run-time. It is more efficient to pre-compute values of the transcendental function ($\cos(\theta_m)$ and $\sin(\theta_m)$) before run-time, and store the values in look-up tables (not shown) in the memory **155** (FIG. **1B**). However, since in one embodiment of the invention a number of candidate source locations is 4326, a number of frequency bin numbers is 640, and a number of microphone signals is M=16, and since θ_m is a function of a candidate source location, a location of a microphone m, and a frequency bin number k, the look-up tables for $\cos(\theta_m)$ and $\sin(\theta_m)$ include 2·4326·640·16=88,596,480 entries. Using a processor with 16 bit precision per entry and eight bits per byte requires approximately 177 M bytes of memory to store the look-up tables.

To reduce memory requirements of the look-up tables and decrease cost of system hardware, alternate (phase delay) look-up tables are generated according to the present invention. In one embodiment of the invention, the method generates a look-up table $D(r,m)=(512\cdot\theta_m)/(2\pi k)=512\cdot b\cdot\Delta_m.v$, where r is a vector from a microphone array to a candidate source location (see FIG. **2A**), and m is the microphone index. If there are 4326 candidate source locations and 16 microphones, then the look-up table D(r,m) has 4326·16=69,216 entries. In addition, the method generates a modulo cosine table cos_table(i)=cos(π·i/256) with 512 entries, where i=0, . . . , 511. Finally, $\cos(\theta_m)$ may be obtained for a given candidate source location and a given frequency bin number k by a formula $\cos(\theta_m)$=cos¯table (0x1FF & int(k·D(r,m))). The argument int(k·D(r,m)) is a product k·D(r,m) rounded to the nearest integer, the argument 0x1FF is a hexadecimal representation of the decimal number 511, and & is a binary "and" function. For example, a binary representation of 0x1FF is the 9-bit representation 1 1 1 1 1 1 1 1 1 . If, for example, $\theta_m=\pi/2$, then int(k·D(r, m))=int((512·θ_m)/2π)=128=0 1 0 0 0 0 0 0 0 in binary. Therefore, $\cos(\theta_m)$=cos_table((1 1 1 1 1 1 1 1 1) & (0 1 0 0 0 0 0 0 0))=$\cos_{_table(}$128)=cos(π·128/256)=cos(π/2).

According to one embodiment of the present invention which comprises 4326 candidate source locations and 16 microphones, the method of generating $\cos(\theta_m)$ and $\sin(\theta_m)$ of the transcendental function $e^{i\theta}{}_m$ requires only three look-up tables: the look-up table D(r,m) with 69,216 entries, the modulo cosine table cos_table(i) with 512 entries, and a modulo sine table sin_table(i) with 512 entries, where the modulo sine table $\sin_{13}$ table(i)=sin(π·i/256). Thus, a total number of 70,240 entries are associated with the three look-up tables, requiring approximately 140 k bytes of memory. The 140 k bytes of memory required for the three tables is more than 1000 times less than the 177 M bytes of memory required to store every value of the transcendental function.

The invention has been explained above with reference to preferred embodiments. Other embodiments will be apparent to those skilled in the art in light of this disclosure. The present invention may readily be implemented using configurations other than those described in the preferred embodiments above. Additionally, the present invention may effectively be used in conjunction with systems other than the one described above as the preferred embodiment. Therefore, these and other variations upon the preferred embodiments are intended to be covered by the present invention, which is limited only by the appended claims.

What is claimed is:

1. A method for computing a location of an acoustic source, comprising the steps of:

receiving acoustic signals from the acoustic source by an array of M-**1** microphones and a reference microphone, each microphone identified by an integer microphone index m, $0 \leqq m \leqq \mathbf{M-1}$;

storing phase-delay look-up tables, the phase-delay look-up tables based upon a plurality of candidate source locations and a spatial configuration of the array of microphones; and

processing the received acoustic signals using the phase-delay look-up tables to compute the location of the acoustic source.

2. The method of claim **1**, wherein an entry in a first phase-delay look-up table is defined by an algebraic expression $D(r,m)=512 \cdot b \cdot \Delta m \cdot v$, where r is a vector to a candidate source location of the plurality of candidate source locations, b is a frequency width, v is inversely proportional to a speed of sound, and Δm is a distance between a location of a microphone m and the candidate source location minus a distance between a location of the reference microphone and the candidate source location.

3. The method of claim **1**, wherein an entry in a second phase-delay look-up table is defined by an algebraic expression $\cos\_table(j)=\cos(\pi \cdot j/256)$, where j is an integer index and 0≦j511.

4. The method of claim **1**, wherein an entry in a third phase-delay look-up table is defined by an algebraic expression $\sin\_table(j)=\sin(\pi \cdot j/256)$, where j is an integer index and $0 \leqq j \leqq 511$.

5. The method of claim **1**, wherein the processing further comprises the steps of:

processing each received acoustic signal to generate blocks of complex coefficients sampled in frequency, each complex coefficient of a block associated with a frequency bin number k, where $0 \leqq k \leqq \mathbf{N-1}$;

computing signal energies, a signal energy received by the array of microphones from a candidate source location of the plurality of candidate source locations for the frequency bin number k determined by multiplying a complex coefficient of a selected block from each received acoustic signal associated with a microphone m by an appropriate phase delay, summing the phase-delayed complex coefficients, and squaring the summation; and

computing the location of the acoustic source by normalizing and summing the signal energies over the N frequency bin numbers for each candidate source location of the plurality of candidate source locations to give a total signal energy for each candidate source location.

6. The method of claim **5**, wherein the processing further comprises the steps of:

digitizing each received acoustic signal;

segmenting each digitized signal into a plurality of blocks, each block of the plurality of blocks including N digital samples Xpm(n), each digital sample Xpm(n) identified by the integer microphone index m, an integer block index p, and an integer sample index n, where $0 \leqq n \leqq \mathbf{N-1}$ and

performing a discrete Fast Fourier Transform (FFT) on each block to transform the N digital samples per block to N complex coefficients Fpm(k) per block, where $0 \leqq k \leqq \mathbf{N-1}$.

7. The method of claim **5**, wherein the appropriate phase delay for the frequency bin number k, the candidate source location, and the microphone m is determined from a first phase-delay look-up table D(r,m), a second phase-delay look-up table cos_table(j), and a third phase-delay look-up table sin_table(j), where r is a vector to the candidate source location of the plurality of candidate source locations and j is an integer computed from an algebraic expression based upon the first look-up table and the frequency bin number k.

8. The method of claim **5**, wherein the computing the location of the acoustic source further comprises the step of determining a maximum total signal energy.

9. A method for computing a location of an acoustic source, comprising the steps of:

receiving analog signals from **M-1** microphones and a reference microphone, each received analog signal and each microphone identified by an integer microphone index m, $0 \leqq m \leqq \mathbf{M-1}$;

digitizing each received analog signal to generate a plurality of digital samples;

segmenting each digitized signal into a plurality of blocks, each block of the plurality of blocks including N digital samples of the plurality of digital samples and each digital sample of the N digital samples identified by the integer microphone index m, an integer block index p, and an integer sample index n, $0 \leqq n \leqq \mathbf{N-1}$;

performing a discrete Fast Fourier Transform (FFT) on each block to transform the N digital samples to N complex coefficients, a complex coefficient Fpm(k) of the N complex coefficients identified by the integer microphone index in, the integer block index p, and an integer frequency bin number k, $0 \leqq k \leqq \mathbf{N-1}$;

searching P blocks of each digitized signal for a maximum signal energy associated with the integer frequency bin number k, identifying a block p' containing the maximum signal energy, $0 \leqq p' \leqq \mathbf{P-1}$;

comparing the maximum signal energy with a threshold energy Et(k), and if the maximum signal energy is less than the threshold energy, setting each complex coefficient of the P blocks of each digitized signal associated with the integer frequency bin number k equal to zero;

determining a plurality of phase delays using look-up tables;

multiplying each complex coefficient by a phase delay eiθm from the plurality of phase delays to generate phase-delayed complex coefficients and summing the phase-delayed complex coefficients over the integer microphone index m for a candidate source location (x,y,z) of a plurality of candidate source locations and for the integer frequency bin number k according to a first algebraic expression

$$Gx,\ y,\ z(k) = \sum_{m=0}^{M-1} ei\theta m F_{m}^{p'}(k);$$

computing a normalized total signal energy for the candidate source location (x,y,z) according to a second algebraic expression;

$$W(x,\ y,\ z) = \sum_{k=kl}^{k=kh} [|Gx,\ y,\ z(k)|2/|S(k)|2],$$

where $0 \leqq k1 \leqq kh \leqq \mathbf{N-1}$ and S(k) is an approximate measure of signal strength for the integer frequency bin number k; and

determining the location of the acoustic source based upon the normalized total signal energies computed for the plurality of candidate source locations.

10. The method of claim **9**, wherein M=16.

11. The method of claim **9**, wherein N=640.

12. The method of claim **9**, wherein P=5.

13. The method of claim **9**, wherein an entry in a first look-up table is defined by an algebraic expression D(r,m)=512·b·Δm·v, where r is a vector to the candidate source location (x,y,z) of the plurality of candidate source locations, b is a frequency width of the integer frequency bin number k, v is inversely proportional to a speed of sound, and Δm is a distance between a location of a microphone m and the candidate source location (x,y,z) minus a distance between a location of the reference microphone and the candidate source location (x,y,z).

14. The method of claim **13**, wherein an integer index j is defined by a third algebraic expression j=0x1FF & int(k·D(r,m)), where int(k·D(r,m)) is a product k·D(r,m) rounded to a nearest integer, 0 x1FF is a hexadecimal representation of a decimal number 511, and & is a binary "and" function.

15. The method of claim **14**, wherein the phase delay eiθm is defined by a fourth algebraic expression eiθm=cos_table(j)+i·sin_table(j), where i is (−1)1/2, cos_table(j)=cos(π·j/256), and sin_table(j)=sin(π·j/256).

16. The method of claim **9**, wherein |S(k)|2 is defined by a fifth algebraic expression

$$|S(k)|2 = \sum_{m=0}^{M-1} |Fp'm(k)|2.$$

17. The method of claim **9**, wherein determining the location of the acoustic source further comprises the step of determining a maximum normalized total signal energy from the normalized total signal energies.

18. An electronic-readable medium having embodied thereon a program, the program being executable by a machine to perform method steps for computing a location of an acoustic source, the method steps comprising:

- receiving acoustic signals from the acoustic source by an array of M-**1** microphones and a reference microphone, each microphone identified by an integer microphone index m, 0≦m≦M-**1**;
- storing phase-delay look-up tables, the phase-delay look-up tables based upon a plurality of candidate source locations and a spatial configuration of the array of microphones; and
- processing the received acoustic signals using the phase-delay look-up tables to compute the location of the acoustic source.

19. The electronic-readable medium of claim **18**, further comprising the steps of:

- processing each received acoustic signal to generate blocks of complex coefficients sampled in frequency, each complex coefficient of a block associated with a frequency bin number k, where 0≦k≦N-**1**;
- computing signal energies, a signal energy received by the array of microphones from a candidate source location of the plurality of candidate source locations for the frequency bin number k determined by multiplying a complex coefficient of a selected block from each received acoustic signal associated with a microphone m by an appropriate phase delay, summing the phase-delayed complex coefficients, and squaring the summation; and
- computing the location of the acoustic source by normalizing and summing the signal energies over the N frequency bin numbers for each candidate source location of the plurality of candidate source locations to give a total signal energy for each candidate source location.

20. The electronic-readable medium of claim **19**, wherein the appropriate phase delay for the frequency bin number k, the candidate source location, and the microphone m is determined from a first phase-delay look-up table D(r,m), a second phase-delay look-up Table cos_table(j), and a third phase-delay look-up table sin_table(j), where r is a vector to the candidate source location of the plurality of candidate source locations and j is an integer computed from an algebraic expression based upon the first look-up table and the frequency bin number k.

21. The electronic-readable medium of claim **19**, wherein the computing the location of the acoustic source further comprises the step of determining a maximum total signal energy.

22. A system for computing a location of an acoustic source, comprising:

- means for receiving acoustic signals from the acoustic source by an array of M-**1** microphones and a reference microphone, each microphone identified by an integer microphone index m, 0≦m≦M-**1**;
- means for storing phase-delay look-up tables, the phase-delay look-up tables based upon a plurality of candidate source locations and a spatial configuration of the array of microphones; and
- means for processing the received acoustic signals using the phase-delay look-up tables to compute the location of the acoustic source.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

PATENT NO. : 6,912,178 B2
DATED : June 28, 2005
INVENTOR(S) : Peter L. Chu, Michael Kenoyer and Richard Washington

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 11,
Line 22, the formula "$0 \leq j511$" should be changed to -- $0 \leq j \leq 511$ --.

Column 12,
Line 26, the word "in" after "microphone index" should be changed to -- m --.
Line 62, the formula "$0 \leq k\mathbf{1} \leq kh \leq N\text{-}\mathbf{1}$" should be changed to -- $0 \leq k\mathbf{1} < kh \leq N\text{-}\mathbf{1}$ --.

Column 14,
Line 25, the word "Table" should be changed to -- table --.
Line 31, the word "the" before "computing" should be removed.

Signed and Sealed this

Thirtieth Day of August, 2005

Jon W. Dudas

JON W. DUDAS
Director of the United States Patent and Trademark Office